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Date: March 25, 1998
Atty Docket No. 7217/55493

TO THE ASSISTANT COMMISSIONER FOR PATENTS
Washington, D.C. 20231

Sir:

With reference to the filing in the United States Patent and Trademark Office of an application for patent in the name(s) of:

Yuji Maeda and Shuichi Maeda

entitled: VECTOR SEARCH METHOD

___ Small entity status under 37 CFR 1.9(f) is claimed and the amounts shown in parentheses below have been employed.

The following are enclosed:

_X_ Specification

_X_ _6_ Claims(s) (including 1 independent claim(s))

_X_ Unsigned Oath or Declaration, Power of Attorney & Petition

_X_ _8_ Sheet(s) of Drawings

_x_ Our check for **$790.00** calculated as follows:

Basic Fee of $790 ($395) .... $ 790.00

______ Total Claims in excess of 20 at $22 ($11).......$.00

______ Ind. Claims in excess of 3 at $82 ($41)........$.00

_____ Fee of $270 ($135) for Mult. Dep. Claim........$.00

Total Filing Fee **$790.00**

Assignment Recording Fee of $40 .... $.00

_X_ Certified copy of each of the following to substantiate the claim for priority:

Application No.	Filing Date	Country
P09-078615	March 28, 1997	Japan

_X_ Please charge any additional fees required for the filing of this application and any other fees required during the pendency of this application or credit any overpayment to Deposit Account No. 03-3125. A duplicate copy of this letter is enclosed.

Respectfully submitted,

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TITLE OF THE INVENTION

Vector Search Method

BACKGROUND OF THE INVENTION

Field of the Invention

The present invention relates to a vector search method for obtaining an optimal sound source vector in vector quantization in compressing to code a audio signal and an acoustic signal.

Description of the Prior Art

Various coding methods are known for compressing a audio signal and an acoustic signal by utilizing statistic features in the time region and frequency band as well as the hearing sense characteristics. These coding methods can divided into a time region coding, a frequency region coding, an analysis-synthesis coding, and the like.

As the effective coding method for compressing to encode an audio signal and the like, there are known a sine wave analysis coding such as harmonic coding and multiband excitation (MBE) coding as well as sub-band coding (SBC), linear predictive coding (LPC), discrete cosine transform (DCT), modified DCT (MDCT), fast Fourier transform (FFT), and the like.

When coding an audio signal, it is possible to predict a present sample value from a past sample value, utilizing that there is a correlation between adjacent sample values. The

adaptive predictive coding (APC) utilizes this characteristic and carries out a coding of a difference between a predicted value and an input signal, i.e., a prediction residue.

In this adaptive prediction coding, an input signal is fetched in a coding unit in which an audio signal can be regarded as almost stationary, for example, in a frame unit of 20 ms and a linear prediction is carried out according to a prediction coefficient obtained by the linear prediction coding (LPC), so as to obtain a difference between the predicted value and the input signal. This difference is quantized and multiplexed with the prediction coefficient and the quantization step width as auxiliary information, so as to be transmitted in a frame unit.

Next, explanation will be given on the code excited linear prediction (CELP) coding as a representative predictive coding method.

The CELP coding uses a noise dictionary called a codebook from which an optimal noise is selected to express an input audio signal and its number (index) is transmitted. In the CELP coding, a closed loop using the analysis by synthesis (AbS) is employed for vector quantization of a time axis waveform, thus coding a sound source parameter.

Fig. 1 is a block diagram showing a configuration of an essential portion of a coding apparatus for coding an audio signal by using the CELP. Hereinafter, explanation will be given on the CELP coding with reference to the configuration of this coding apparatus.

An audio signal supplied from an input terminal 10 is firstly subjected to the LPC (linear predictive coding) analysis in an LPC analyzer 20, and a prediction coefficient obtained is transmitted to a synthesis filter 30. Moreover, the prediction coefficient is also transmitted to a multiplexer 130.

In the synthesis filter 30, the prediction coefficient from the LPC analyzer 20 is synthesized with signed vectors supplied from an adaptive code book 40 and a noise codebook 60, which will be detailed later, through amplifiers 50 and 70 and an adder 80.

An adder 90 determines a difference between the audio signal supplied from the input terminal 10 and a prediction value from the synthesis filter 30, which is transmitted to a hearing sense weighting block 100.

In the hearing sense weighting block 100, the difference obtained in the adder 90 is weighted, considering the characteristics of the hearing sense of a human. An error calculator 110 searches a signed vector to minimize a distortion of the difference weighted by the hearing sense, i.e., a difference between the prediction value from the synthesis filter 30 and the input audio signal, and gains of the amplifier 50 and 70. The result of this search is transmitted as an index to the adaptive codebook 40, the noise codebook 60, and a gain codebook 120 as well as to the multiplexer 130 so as to be transmitted as a transmission path sign from an output terminal 140.

Thus, an optimal signed vector to express the input audio signal is selected from the adaptive codebook 40 and the noise codebook 60, and the optimal gain is determined for synthesizing them. It should be noted that the aforementioned processing can be carried out after the hearing-sense weighting the audio signal supplied from the input terminal 10, and signed vectors stored in the codebooks may be hearing-sense wieghted.

Next, explanation will be given on the aforementioned adaptive codebook 40, the noise codebook 60, and the gain codebook 120.

In the CELP coding, a sound source vector for expressing an input audio signal is formed as a linear sum of a signed vector stored in the adaptive codebook 40 and s signed vector stored in the noise codebook 60. Here, the indexes of the respective codebooks to express the sound source vector minimizing the hearing-sense weighted difference from the input signal vector are determined by calculating the output vector of the synthesis filter 30 for all the signed vectors stored and calculating errors in the error calculator 110.

Moreover, the gain of the adaptive codebook in the amplifier 50 and the gain of the noise codebook in the amplifier 70 are also coded by way of the similar search.

The noise codebook 60 normally contains a series of vectors of the Gauissian noise with dispersion 1 as the codebook vectors in number 2 powered by the number of bits. And normally, a combination of the codebook vectors is selected

so as to minimize the distortion of the sound source vector obtained by adding an appropriate gain to these codebook vectors.

The quantization distortion when quantizing the selected codebook vectors can be reduced by increasing the number of dimensions of the codebook. For example, the codebook used is in 40 dimensions and 2 powered by 9 (the number of bits), i.e., 512 terms.

By using this CELP coding, it is possible to obtain a comparatively high compression ratio and a preferable sound quality. However, the use of a codebook of a large number of dimensions requires a large calculation amount in the synthesis filter and a large memory amount of the codebook, which makes difficult a real-time processing. If a high sound quality is to be assured, a great delay is caused. Moreover, there is another problem that only one bit error in the code brings about a completely different vector reproduced. That is, such a coding is weak for the sign error.

In order to improve the aforementioned problems of the CELP coding, the vector sum excited linear prediction (VSELP) coding is employed. Hereinafter, this VSELP coding will be explained with reference to Fig. 3 and fig. 3.

Fig. 2 is a block diagram showing a configuration of a noise codebook used in a coding apparatus for coding an audio signal by way of the VSELP.

The VSELP coding employs a noise codebook 260 consisting of a plurality of predetermined basic vectors. Each of the

number M of basic vectors stored in the noise codebook 260 is multiplied by a factor +1 or −1 to reverse the value according to the index decoded with a code additional section 270-1 to 270-M by a decoder 210. The M basic vectors multiplied by the factor +1 or −1 are combined with one another in an adder 280 to create 2^M noise signed vectors.

As a result, by carrying out a convolution calculation for the M basic vectors and addition and subtraction thereof, it is possible to obtain a convolution calculation result for all the noise signed vectors. Moreover, as only the M basic vectors should be stored in the noise codebook 260, it is possible to reduce the memory amount. Besides, it is possible to enhance the durability for a sign error because the 2^M noise signed vectors created has a redundant configuration which can be expressed by addition and subtraction of the basic vectors.

Fig. 3 is a block diagram showing a configuration of an essential portion of a VSELP coding apparatus having the aforementioned noise codebook. In this VSELP coding apparatus, the number of noise codebooks which is normally 512 in the ordinary CELP coding apparatus is reduced to 9, and each of the signed vectors (basic vectors) is added with a sign +1 or −1 by a sign adder 365, so that a linear sum of these is obtained in an adder 370, so as to create $2^9 = 512$ noise signed vectors.

The main feature of the VSELP coding is as has been described above that a noise signed vector is formed as a linear sum of basic vectors and that the gain of the adaptive codebook and the gain of the noise codebook are

vector-quantized at once.

The basic configuration of such a VSELP coding is a coding method of analysis by way of synthesis, i.e., carrying out a linear prediction synthesis of a pitch frequency component and a noise component as the excitation sources. That is, a waveform is selected in vector unit from an adaptive codebook 340 which depends on a pitch frequency of an input audio signal and a noise codebook 360 for carrying out a linear prediction synthesis, so as to select a signed vector and a gain which minimize the difference from the waveform of the input audio signal.

In the VSELP coding, a signed vector from the adaptive codebook expressing the pitch component of an input audio signal and a signed vector from the noise codebook expressing the noise component of the input audio signal are both vector-quantized, so as to simultaneously obtain two optimal parameters in combination.

In this process, as the basic vector has only the freedom of being added by +1 or -1 and the vector of the adaptive codebook is not orthogonal to the basic vector, the coding efficiency is lowered if the CELP procedure is employed to successively determine the vector of the adaptive codebook and the gain of the noise signed vector. To cope with this, in the VSELP, the basic vector sign is determined according to a procedure as follows.

Firstly, the pitch frequency of the input audio signal is searched to determine a signed vector of the adaptive codebook.

Next, the noise basic vector is projected to a space orthogonal to the signed vector of the adaptive codebook and an inner product with the input vector is calculated, so as to determine the signed vector of the noise codebook.

Next, according to the two signed vectors determined, the codebook is searched to determine a combination of a gain β and a gain γ which minimizes the difference between the vector synthesized and the input audio signal. For quantization of the two gains, a pair of two parameters equally converted is used. Here, the β corresponds to a long-term prediction gain coefficient and the γ corresponds to a scalar gain of the signed vector.

Although the calculation amount for the codebook search in the VSELP coding is reduced than the calculation amount in the CELP coding, it is desired to further improve the processing speed, further reducing the delay.

SUMMARY OF THE INVENTION

It is therefore an object of the present invention to simply the codebook search in the vector quantization when coding an audio signal or the like, enabling to improve the vector search speed.

In order to achieve the aforementioned object, in the vector search method according to the present invention wherein among prediction vectors obtained according to synthetic vectors obtained by synthesizing a plurality of basic vectors each multiplied by a factor +1 or −1, such a prediction vector

is determined that makes minimum a difference from a given input vector or makes maximum an inner product with the given input vector, the calculation to obtain the difference from the input vector or the inner product with the input vector is carried out by changing the combinations of the aforementioned factors multiplied for each of the plurality of basic vectors, according to the Gray code, so that an intermediate value Gu obtained from a synthetic vector created according to the Gray code u is expressed by an intermediate value Gi based on i adjacent to the Gray code u and a change DGu between them.

Furthermore, the combination of the basic vectors which makes minimum the difference between the input vector and the prediction vector or makes maximum an inner product between them is obtained by using a difference between a change of the synthetic vector when a predetermined bit position of the Gray code is changed and a change of the synthetic vector when a different bit position is changed.

According to the aforementioned vector search method, by utilizing the characteristic of the Gray code, it is possible to use a calculation result obtained for carrying out the next calculation, thus enabling to increase the vector search speed.

BRIEF DESCRIPTION OF THE DRAWINGS

Fig. 1 is a block diagram showing a configuration example of a coding apparatus for explanation of the CELP coding.

Fig. 2 is a block diagram showing the configuration of the noise codebook used in the VSELP coding.

Fig. 3 is a block diagram showing a configuration example of a coding apparatus for explanation of the VSELP coding.

Fig. 4 shows an example of the binary Gray code.

Fig. 5 is a flowchart showing a procedure of the vector search method according to the present invention.

Fig. 6 shows a calculation amount and a memory write amount in the vector search method according to the present invention in comparison to the conventional vector search.

Fig. 7 explains the PSI-CELP.

Fig. 8 is a block diagram showing a configuration example of a coding apparatus for explanation of the PSI-CELP coding.

DETAILED DESCRIPTION OF PREFERRED EMBODIMENTS

Description will now be directed to the vector search method according to preferred embodiments of the present invention.

Firstly, explanation will be given on a case of vector quantization carried out in the aforementioned VSELP coding apparatus.

In the waveform coding and analysis-synthesis system, instead of quantizing respective sample values of a waveform and spectrum envelope parameters, a plurality of values in combination (vector) are expressed as a whole with a single sign. Such a quantization method is called vector quantization. In the coding by way of waveform vector quantization, a waveform after sampled is cut out for a predetermined time interval as a coding unit and a waveform

pattern during the interval is expressed by a single sign. For this, various waveform patterns are stored in memory in advance and a sign is added to them. The correspondence between the sign and the patterns (signed vector) is indicated by a codebook.

For an audio signal waveform, a comparison is made with each of the parameters stored in the codebook for the respective time intervals and a sign of the waveform having the highest similarity is used to express the waveform of the interval. Thus, various input sounds are expressed with a limited number of patterns. Consequently, appropriate patterns to minimize the entire distortion should be stored in the codebook, considering the pattern distribution and the like.

The vector quantization can be a highly effective coding based on the facts that the patterns to be realized have various specialties such that a correlation can be seen between sample points in a certain interval of an audio waveform and the sample points are smoothly connected.

Next, explanation will given on the vector search for searching a signed vector which minimizes the difference between an input vector and a synthesized vector formed from an optimal combination of a plurality of vectors selected from the codebook.

Firstly, it is assumed that p (n) is an input audio signal weighted with hearing sense and $q'_m(n)$ $(1 \le m \le M)$ is a basic vector orthogonal to a long-term prediction vector weighted with hearing sense.

Expression (1) gives an inner product of the input vector and the synthesized vector formed by a combination of a plurality of vectors selected from the codebook. That is, by obtaining θ_{ij} which makes the Expression (1) maximum, the inner product between the synthesized vector and the input vector becomes maximum.

It should be noted that the combination θ_{ij} is −1 if the bit j of the sign word i is 0, and 1 if the bit j of the sign word i is 1 ($0 \leq i \leq 2^{M} - 1$, $1 \leq m \leq M$).

[Expression 1]

$$\frac{\left(\sum_{n=0}^{N-1}\sum_{m=1}^{M}\theta_{im}q_m^{\prime}(n)p(n)\right)^2}{\sum_{n=0}^{N-1}\left(\sum_{m=1}^{M}\theta_{im}q_m^{\prime}(n)\right)^2} \rightarrow Max. \qquad \ldots (1)$$

The denominator of the Expression (1) can be developed to obtain Expression (2).

[Expression 2]

$$2\sum_{n=0}^{N-1}\sum_{j=2}^{M}\sum_{m=1}^{j-1}\theta_{im}\theta_{ij}q_m^{\prime}(n)q_j^{\prime}(n)+\sum_{n=0}^{N-1}\sum_{m=1}^{M}q_m^{\prime}(n)^2 \qquad \ldots (2)$$

Here, a variable R_m given by Expression (3) and a variable D_{mj} given by Expression (4) are introduced.

[Expression 3]

$$R_m = 2\sum_{n=0}^{N-1} q_m'(n)p(n) \qquad \ldots (3)$$

$$D_{mj} = 4\sum_{n=0}^{N-1} q_m'(n)q_j'(n) \qquad \ldots (4)$$

These variables R_m and D_{mj} are introduced into Expression (1) to obtain Expression (5).

[Expression 4]

$$\frac{\left(\frac{1}{2}\sum_{m=1}^{M} \theta_{im} R_m\right)^2}{\frac{1}{2}\sum_{j=2}^{M}\sum_{m=1}^{j-1} \theta_{im}\theta_{ij} D_{mj} + \frac{1}{4}\sum_{m=1}^{M} D_{mm}} \qquad \ldots (5)$$

Here, a variable C_i given by Expression (6) and a variable G_i given by Expression (7) are further introduced.

[Expression 5]

$$C_i = \frac{1}{2}\sum_{m=1}^{M} \theta_{im} R_m \qquad \ldots (6)$$

$$G_i = \frac{1}{2}\sum_{j=2}^{M}\sum_{m=1}^{j-1} \theta_{im}\theta_{ij} D_{mj} + \frac{1}{4}\sum_{m=1}^{M} D_{mm} \qquad \ldots (7)$$

By using these variables C_i and G_i, Expression (1) can be rewritten into Expression (8). That is, by obtaining the variables C_i and G_i to maximize the Expression (8), it is possible to make maximum the correlation between the synthesized vector and the input vector.

$$C_i^2/G_i - \text{Max.} \qquad (8)$$

By the way, if there is a sign word u which is different from the sign word i only in the bit position v, and if C_i and G_i are known, then C_u and G_u can be expressed by Expressions (9) and (10).

[Expression 6]

$$Cu = C_i + \theta_{uv} R_v \qquad \ldots (9)$$

$$G_u = G_i + \sum_{j=1}^{v-1} \theta_{uj}\theta_{uv} D_{jv} + \sum_{j=v+1}^{M} \theta_{uj}\theta_{uv} D_{jv} \qquad \ldots (10)$$

By utilizing this and by converting the sign word i by using the binary Gray code, it is possible to calculate with a high efficiency the optimal combination of a plurality of signed vectors selected from the codebook. Note that the Gray code will be detailed later.

The Expression (10) can be rewritten into Expression (11) if ΔG_u is assumed to be a change from G_1 to Gu.

[Expression 7]

$$\Delta G_u = \sum_{j=1}^{v-1} \theta_{uj}\theta_{uv}D_{jv} + \sum_{j=v-1}^{M} \theta_{uj}\theta_{uv}D_{jv} \qquad \ldots (11)$$

Here, the sign word u' of the binary Gray code differs from the sign word i only in the bit position V. The sign word u' differs from the preceding sign word u only in one bit other than the bit position v.

Now, if w is assumed to be the aforementioned bit position, the sign of θ_{uv} is reversed and the relationship of Expression (12) can be obtained from the Expression (11).

$$\Delta G_{u'} = -\Delta G_u + 2\theta_{uw}\theta_{uv}D_{wv} \qquad (12)$$

From this, it is possible to use the Expression (11) to obtain the change ΔG_u when the bit position V has changed firstly in the binary Gray code and the Expression (12) to obtain the change at the same bit position V after that, thus enhancing the vector search speed.

Fig. 4 shows the binary Gray code when M = 4. As shown here, the Gray code is a kind of cyclic code in which two adjacent sign words differ from each other only in one bit.

Here, if an attention is paid on the bit position V = 3, for example, the value is changed when N changes from 3 to 4 as indicated by a reference numeral 425 and when N changes from 11 to 12 as indicated by a reference numeral 426. That is, if the Gray code when N = 4 is compared to the Gray code when N = 12, the only difference in the bit w (W = 4), excluding the bit v (V = 3).

Here, if it is assumed that the Gray code when N = 4 is u, and the Gray code when N = 12 is u', then

$$\text{When } N = 4\text{:}\quad \theta_{u1} = -1,\ \theta_{u2} = 1,\ \theta_{u3} = 1,\ \theta_{u4} = -1$$
$$\text{When } N = 12\text{:}\quad \theta_{u'1} = -1,\ \theta_{u'2} = 1,\ \theta_{u'3} = -1,\ \theta_{u'4} = 1 \qquad (13)$$

From this and the Expression (11), the following can be obtained.

$$\text{When } N = 4\text{:}\quad \Delta G_u = \theta_{u3}\,\{\theta_{u1}D_{13} + \theta_{u2}D_{23} + \theta_{u4}D_{43}\}$$
$$\text{When } N = 12\text{:}\quad \Delta G_{u'} = \theta_{u'3}\,\{\theta_{u'1}D_{13} + \theta_{u'2}D_{23} + \theta_{u'4}D_{43}\} \qquad (14)$$

As has been described above, because the bit position V = 1 and 2 are with an identical sign and the bit position V = 3 and 4 are with different signs, the following are satisfied.

$$\Delta G_{u'} = -\theta_{u3}\ \{\theta_{u1}D_{13} + \theta_{u2}D_{23} + (-\theta_{u4})D_{43}\} \quad (15a)$$
$$= -\theta_{u3}\ \{\theta_{u1}D_{13} + \theta_{u2}D_{23} + \theta_{u4}D_{43}\} + 2\theta_{u3}\theta_{u4}D_{43}$$
$$= -\Delta G_u + 2\theta_{u3}\theta_{u4}D_{43} \quad (15b)$$

That is, the Expression (15a) can be simplified into the Expression (15b).

Fig. 5 is a flowchart showing the aforementioned procedure of the vector search method according to the present invention.

Firstly, in step ST1, the variable R_m is calculated from the Expression (3), and the variable D_{mj}, from the Expression (4).

In step ST2, the variable C_0 is calculated from the Expression (6), and the variable G_0, from the Expression (7).

In step ST3, C_i ($1 \leq i \leq 2M - 1$) is calculated from the Expression (9).

In step ST4, the bit V-1 is calculated.

In step ST5, the change amount ΔG_u of G_u when a certain bit V firstly changes is calculated from the Expression (11).

In step ST6, the ΔG_u when the remaining bit V changes is calculated from the Expression (12).

In step ST7, the bit V is set to V + 1.

In step ST8, it is determined whether the V is equal to or less than M. If V is equal to or less than M, control is returned to step ST5 to repeat the aforementioned procedure. On the other hand, if V is greater than M, control is passed to step ST6.

In step ST9, $G_u = G_1 + \Delta G_u$ (wherein $1 \leq u \leq 2M - 1$) is

calculated, completing the vector search.

Fig. 6 shows the G_1 calculation processing amount obtained by the vector search method according to the present invention in comparison to the processing of the conventional vector search method.

Fig. 6A shows the comparison result in the number of calculations for multiplication. Moreover, Fig. 6B shows the comparison results in the number of calculations for the addition and subtraction. From these results, it can be seen the effect that as the M increases, the number of calculations is reduced.

Moreover, Fig. 6C shows the comparison result in the number of writing times into memory. This result shows that the number of writing times into memory is increased twice in comparison to the conventional vector search method, regardless of the M value.

Next, explanation will be given on the vector search method according to an embodiment of the present invention employed in the vector quantization in the PSI-CELP coding.

The PSI-CELP (pitch synchronous innovation CELP) coding is a highly effective audio coding for obtaining an improved sound quality for the sound-existing portion by periodicity processing of signed vectors from the noise codebook with a pitch periodicity (pitch lag) of the adaptive codebook.

Fig. 7 schematically shows the periodicity processing of the pitch of a signed vector from the noise codebook. In the aforementioned CELP coding, adaptive codebook is used for

effectively expressing an audio signal containing a periodic pitch component. However, when the bit rate is lowered to the order of 4 kbs, the number of bits assigned for the sound source coding is decreased and it becomes impossible to sufficiently express the audio signal containing a periodic pitch component with the adaptive codebook alone.

To cope with this, in the PSI-CELP coding system, the pitch of the signed vector from the noise codebook is subjected to periodicity processing. This enables to accurately express the audio signal containing a periodic pitch component which cannot be sufficiently expressed by the adaptive codebook alone. It should be noted that the lag (pitch lag) L represents a pitch cycle expressed in the number of samples.

Fig. 8 is a block diagram showing a configuration example of an essential portion of a PSI-CELP coding apparatus. Hereinafter, explanation will be given on this PSI-CELP coding with reference to Fig. 8.

The PSI-CELP coding is characterized by carrying out the pitch periodicity processing of the noise codebook. This periodicity processing is to deform an audio signal by taking out only a pitch periodic component which is a basic cycle of the audio signal so as to be repeated.

An audio signal supplied from an input terminal 710 is firstly subjected to a linear prediction analysis in a linear prediction analyzer 720 and a prediction coefficient obtained is fed to a linear prediction synthesis filter 730. In the synthesis filter 730 the prediction coefficient from the LPC

analyzer 720 is synthesized with signed vectors supplied from an adaptive codebook 640 and noise codebooks 680, 760, and 761 respectively via amplifiers 650 and 770 and an adder 780.

The noise signed vector from the noise codebook 660 is a vector selected from 32 basic vectors by a selector 655 and multiplied by a factor +1 or −1 by a sign adder 657. The noise signed vector multiplied by the factor +1 or −1 and the signed vector from the adaptive codebook 640 are selected by a selector 652 and added with a predetermined gain g0 by the amplifier 650 so as to be supplied to the adder 780.

On the other hand, the noise signed vectors from the noise codebooks 760 and 761 are selected respectively from 16 basic vectors by selectors 755 and 756 and subjected to pitch periodicity processing by pitch cyclers 750 and 751, after which they are multiplied by a factor +1 or −1 by sign adders 740 and 741 so as to be supplied to an adder 765. After this, they are given a predetermined gain g_1 in the amplifier 770 and supplied to the adder 780.

The signed vectors which have been given a gain respectively by the amplifiers 650 and 770 are added in the adder 780 and supplied to the linear prediction synthesis filter 730.

In an adder 790, a difference is obtained between the audio signal supplied from the input terminal 710 and the prediction value from the linear prediction synthesis filter 730.

In a hearing sense weighting distortion minimizer 800, the

difference obtained by the adder 790 is subjected to hearing sense weighting, considering the human hearing sense characteristics. The difference weighted with the hearing sense, i.e., a signed vector and a gain are determined to minimize a difference error between the prediction value from the linear prediction synthesis filter 730 and the input audio signal. The results are transmitted as an index to the adaptive codebook 640, the noise codebooks 660, 760, and 761, and outputted as a transmission path sign.

By the way, in the LSP middle band second stage quatization, the Expression (16) gives a Euclid distance between the synthesized vector made from a combination of a plurality of vectors selected from codebooks and the input middle band LSP error vector. That is, this calculation is carried out by obtaining a pair θ(k, i) which minimizes the Euclid distance $D(k)^2$ given by the Expression (16), wherein it is assumed that $0 \le k \le MM - 1$ and $0 \le i \le 7$.

[Expression 8]

$$D(k)^2 = \sum_{j=0}^{7} \left(lspe(k, j) - \sum_{i=0}^{7} \theta(k, i) C_{LSPM2}(i, j) \right)^2 \qquad \ldots (16)$$

This Expression (16) is developed into Expression (17) as follows.

[Expression 9]

$$D(k)^2 = \sum_{j=0}^{7} lspe(k, j)^2 - 2\sum_{i=0}^{7} \theta(k, i) \sum_{j=0}^{7} lspe(k, j)\, C_{LSPM2}(i, j)$$

$$+2\sum_{i=0}^{7} \sum_{m=i+1}^{7} \theta(k, i)\,\theta(k, m) \sum_{j=0}^{7} C_{LSPM2}(i, j)\, C_{LSPM2}(m, j) \qquad \ldots (17)$$

$$+\sum_{i=0}^{7} \sum_{j=0}^{7} C_{LSPM2}(i, j)^2$$

Here, a variable R(k, i) $(o < k < MM - i, 0 < 1 < 7)$ given by Expression (18) and a variable D (i, m) $(0 < i, m < 7)$ given by Expression (19) are introduced.

[Expression 10]

$$R(k, i) = 2\sum_{j=0}^{7} lspe(k, j)\, C_{LSPM2}(i, j) \qquad \ldots (18)$$

$$D(i, m) = 4\sum_{j=0}^{7} C_{LSPM2}(i, j)\, C_{LSPM2}(m, j) \qquad \ldots (19)$$

In the Expression (17), the first term of the right side is always constant and accordingly can be ignored. By substituting the aforementioned variables R and D, it is necessary to obtain θ(k, i) which satisfies the relationship defined by Expression (20) as follows.

[Expression 11]

$$-\sum_{i=0}^{7} \theta(k,\ i)R(k,\ i)+\frac{1}{2}\sum_{i=0}^{7}\sum_{m=i-1}^{7} \theta(k,\ i)\theta(k,\ m)D(i,\ m) + \frac{1}{4}\sum_{i=0}^{7} D(i,\ i) \rightarrow Min. \qquad \ldots (20)$$

Here, a variable C_I given by Expression (21) and a variable G_I given by Expression (22) are further introduced (wherein $0 \le I \le 2^8 - 1$).

[Expression 12]

$$C_I = \frac{1}{2}\sum_{i=0}^{7} \theta(k,\ i)R(k,\ i) \qquad \ldots (21)$$

$$G_I = \frac{1}{2}\sum_{i=0}^{7}\sum_{m=i+1}^{7} \theta(k,\ i)\theta(k,\ m)D(i,\ m)+\frac{1}{4}\sum_{i=0}^{7} D(i,\ i) \qquad \ldots (22)$$

The aforementioned variables CI and GI are introduced into the Expression (20) to obtain the following.

$$-2*C_I + G_I - Min. \qquad (23)$$

That is, it is possible to minimize the error by obtaining the variables C_I and G_I which minimize the Expression (23).

In the aforementioned vector search in the PSI-CELP coding system, Expressions (21) and (22) have identical forms as the

Expressions (9) and (10) in the aforementioned vector search in the VSELP coding. Consequently, the aforementioned vector search method according to the present invention can also be applied to the PSI-CELP, enhancing the vector search speed.

The vector search method according to the present invention, utilizing the Gray code characteristic, uses a result of a calculation which has been complete, for carrying out the next calculation, thus enabling to simplify the calculation of the synthesized vector and increase the vector search speed.

WHAT IS CLAIMED IS

1. A vector search method in which a difference error between a prediction vector and an input vector is calculated in such a way that combinations of factors respectively multiplied by a plurality of basic vectors are changed according to the Gray code.

2. A vector search method as claimed in Claim 1, wherein an intermediate value Gu obtained by calculation of a synthetic vector created according to a sign word u of the Gray code is expressed by an intermediate value Gi obtained by a calculation of a synthetic vector created according to an adjacent sign word i different from said sign word u only in a predetermined bit position v and a change ΔGu calculated by utilizing the Gray code characteristic, and

said ΔGu is used to express a change ΔGu' between an intermediate value Gi' according to another sign word i' in said Gray code and an intermediate value Gu' according to an adjacent sign word u' different from said sign word i' only in a predetermined bit position v.

3. A vector search method as claimed in Claim 2, wherein said prediction vector is created through a prediction synthesis filter by synthesizing said synthetic vector and a vector based on a past sound source signal.

4. A vector search method as claimed in Claim 2, wherein

said sign word u' in said Gray code differs from said sign word u only in one bit position w excluding the predetermined bit position v, and

said change ΔGu' is expressed as a sum of said change ΔGu already obtained according to said sign word u of said Gray code and a difference between said change ΔGu and said ΔGu'.

5. A vector search method as claimed in Claim 2, wherein the calculation to minimize the difference between said prediction vector and said input vector is a calculation to determine such a synthetic vector from synthetic vectors created by synthesizing basic vectors for the sign word i of the Gray code that makes maximum an inner product with said input vector, and

said inner product is expressed, by using two variables Ci and Gi, as Ci2/Gi, whose value is made maximum.

6. A vector search method as claimed in Claim 2, wherein the calculation to minimize the difference between said prediction vector and said input vector is a calculation to determine such synthetic vector from synthetic vectors created by synthesizing basic vectors for the sign word i of the Gray code that makes minimum an Euclid distance from said input vector, and

said Euclid distance is expressed by a sum of two variables Ci and Gi, which sum is minimized.

ABSTRACT

The present invention simplifies a codebook search in the vector quantization when coding an audio signal or the like, enhancing the vector search speed.

Each of the M basic vectors in a noise code book 260 is multiplied by a factor ±1 in a sign adder 270 and combined in an adder 280 to create 2^M noise signed vectors. Here, the characteristic of the binary Gray code is utilized as follows. A change ΔG_u obtained between a noise signed vector based on a signed word i of the binary Gray code and a noise signed vector based on a sign word u adjacent to the sign ward i and different from the sign word i only in a predetermined bit position v is used in such a manner that a sign word u' which is next to reverse the bit position v on the Gray code sequence can express a change $\Delta G_{u'}$ from the noise signed vector by utilizing the fact that the sign word u' differs from the sign word u only in one bit position w excluding the bit position V. Thus, calculation is simplified, increasing the vector search speed.

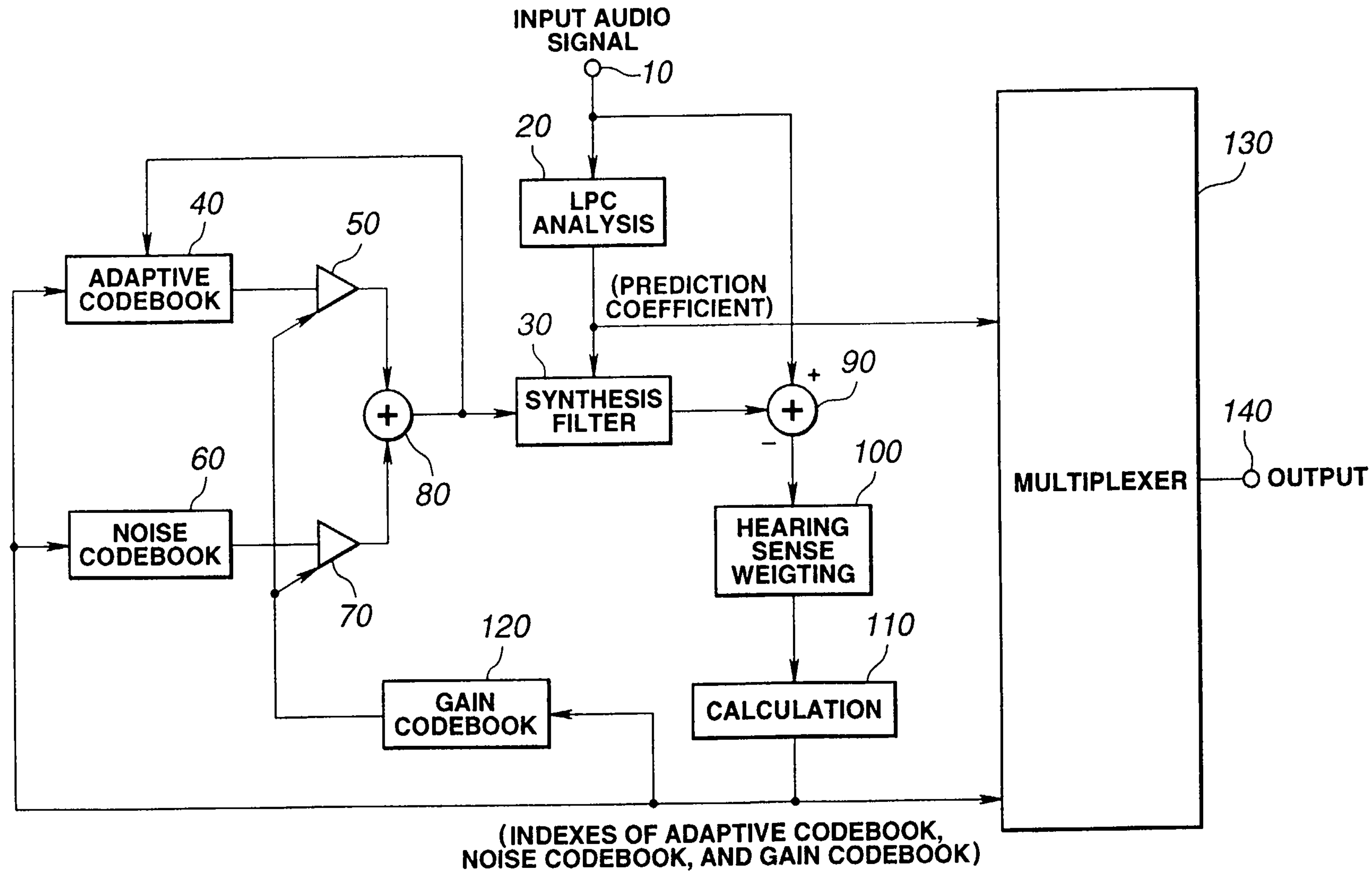
INPUT AUDIO SIGNAL
10
20
LPC ANALYSIS
(PREDICTION COEFFICIENT)
30
SYNTHESIS FILTER
40
ADAPTIVE CODEBOOK
50
60
NOISE CODEBOOK
70
80
90
+
−
100
HEARING SENSE WEIGTING
110
CALCULATION
120
GAIN CODEBOOK
130
MULTIPLEXER
140
OUTPUT
(INDEXES OF ADAPTIVE CODEBOOK, NOISE CODEBOOK, AND GAIN CODEBOOK)

FIG.1

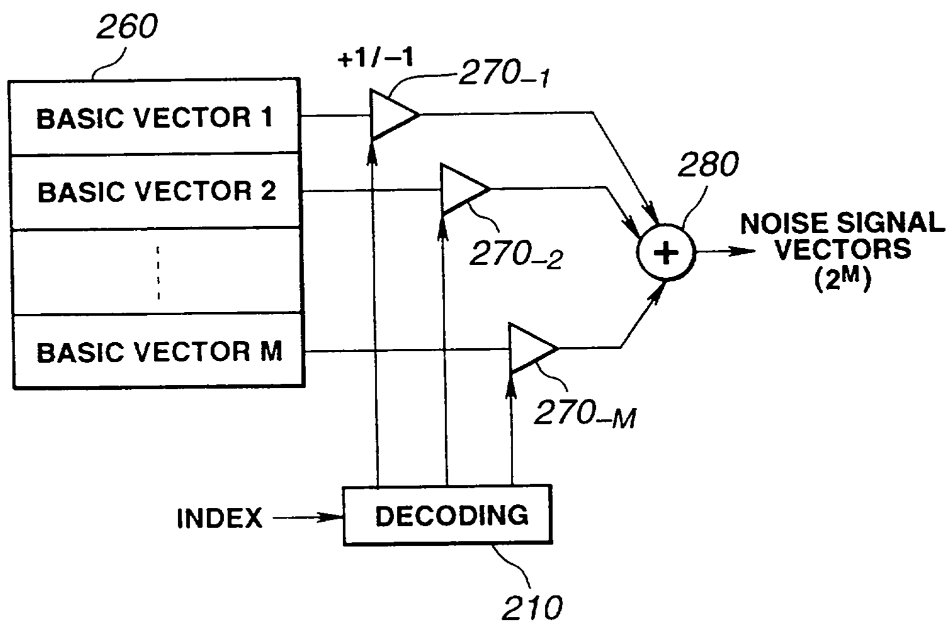
260
BASIC VECTOR 1
BASIC VECTOR 2
BASIC VECTOR M
+1/−1
270-1
270-2
270-M
280
NOISE SIGNAL VECTORS (2^M)
INDEX
DECODING
210

FIG.2

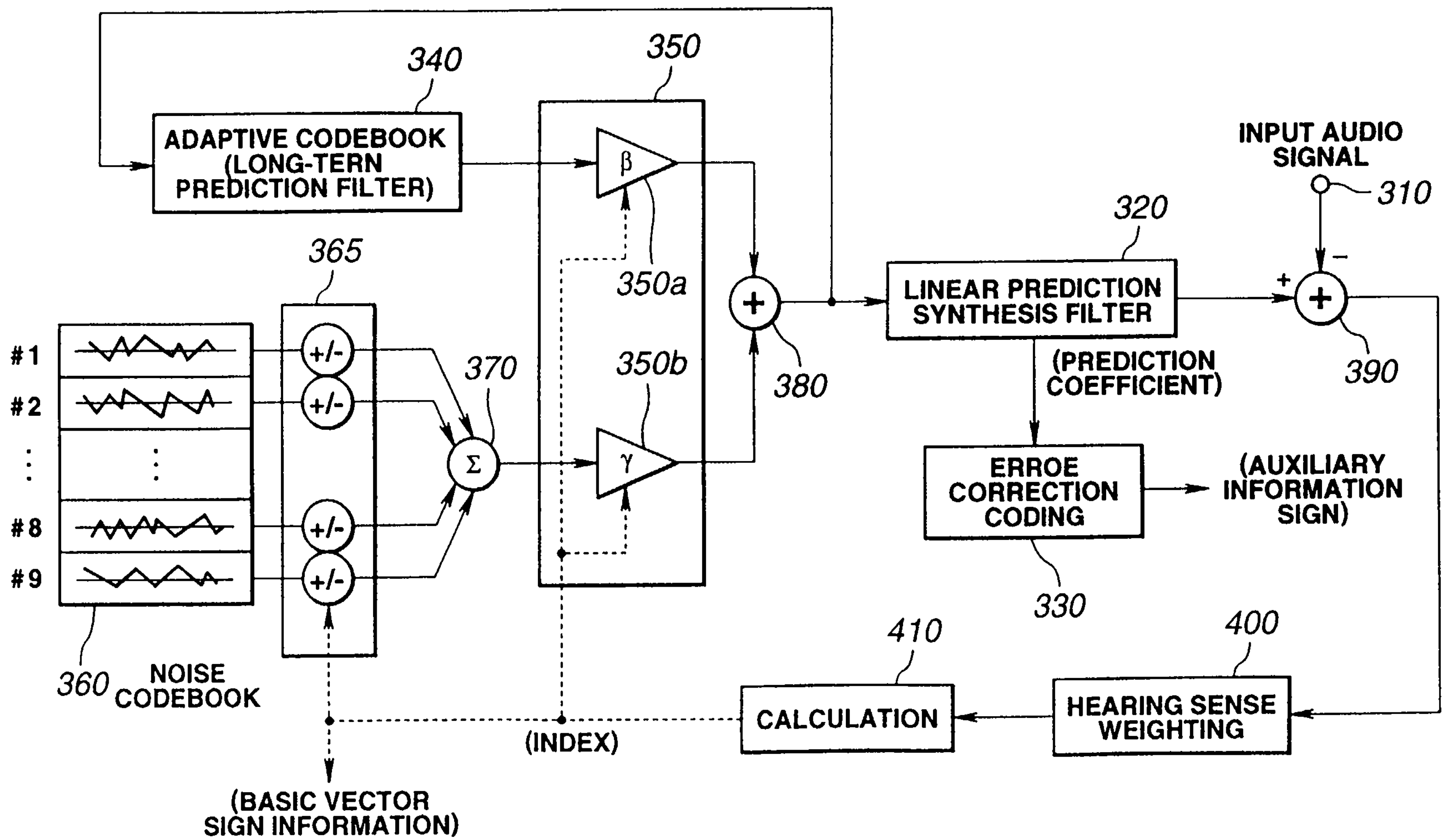
ADAPTIVE CODEBOOK (LONG-TERN PREDICTION FILTER)
340
350
350a
350b
β
γ
365
+/-
370
Σ
380
#1
#2
#8
#9
360
NOISE CODEBOOK
LINEAR PREDICTION SYNTHESIS FILTER
320
INPUT AUDIO SIGNAL
310
390
+
−
(PREDICTION COEFFICIENT)
ERROE CORRECTION CODING
330
(AUXILIARY INFORMATION SIGN)
HEARING SENSE WEIGHTING
400
CALCULATION
410
(INDEX)
(BASIC VECTOR SIGN INFORMATION)

FIG.3

N	BIT V 4 3 2 1
0	0 0 0 0
1	0 0 0 1
2	0 0 1 1
3	0 0 1 0
4	0 1 1 0
5	0 1 1 1
6	0 1 0 1
7	0 1 0 0
8	1 1 0 0
9	1 1 0 1
10	1 1 1 1
11	1 1 1 0
12	1 0 1 0
13	1 0 1 1
14	1 0 0 1
15	1 0 0 0

425

426

FIG.4

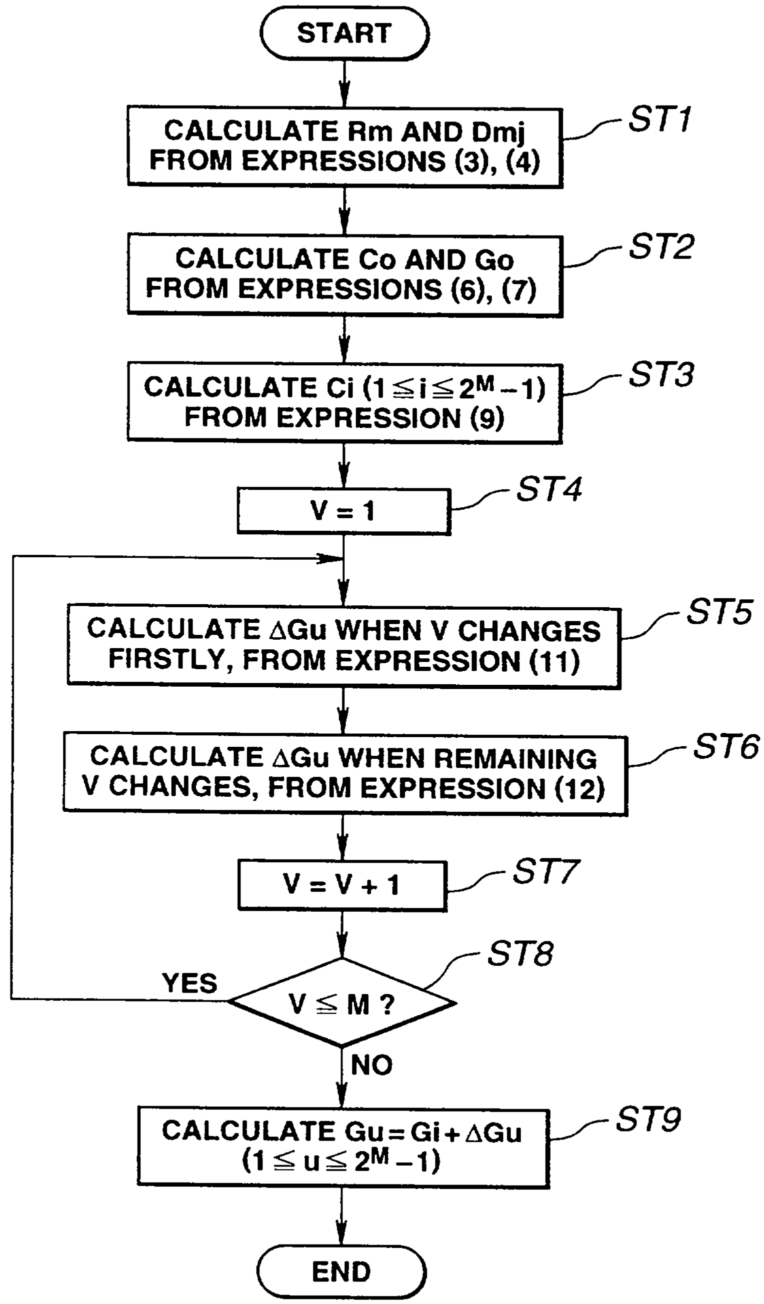
START
CALCULATE Rm AND Dmj FROM EXPRESSIONS (3), (4)
ST1
CALCULATE Co AND Go FROM EXPRESSIONS (6), (7)
ST2
CALCULATE Ci ($1 \leqq i \leqq 2^M - 1$) FROM EXPRESSION (9)
ST3
V = 1
ST4
CALCULATE ΔGu WHEN V CHANGES FIRSTLY, FROM EXPRESSION (11)
ST5
CALCULATE ΔGu WHEN REMAINING V CHANGES, FROM EXPRESSION (12)
ST6
V = V + 1
ST7
$V \leqq M$?
ST8
YES
NO
CALCULATE Gu = Gi + ΔGu ($1 \leqq u \leqq 2^M - 1$)
ST9
END

FIG.5

FIG.6A

M	MULTIPLICATION		
	PRESENT INVENTION	CONVENTIONAL	RATIO
4	57	90	0.633
5	118	248	0.475
6	231	630	0.367
7	444	1524	0.291
8	853	3570	0.239

FIG.6B

M	ADDITION AND SUBTRACTION		
	PRESENT INVENTION	CONVENTIONAL	RATIO
4	49	45	1.089
5	103	124	0.831
6	207	315	0.657
7	409	762	0.537
8	805	1785	0.451

FIG.6C

M	WRITING TO MEMORY		
	PRESENT INVENTION	CONVENTIONAL	RATIO
4	30	15	2.0
5	62	31	2.0
6	126	63	2.0
7	254	127	2.0
8	510	255	2.0

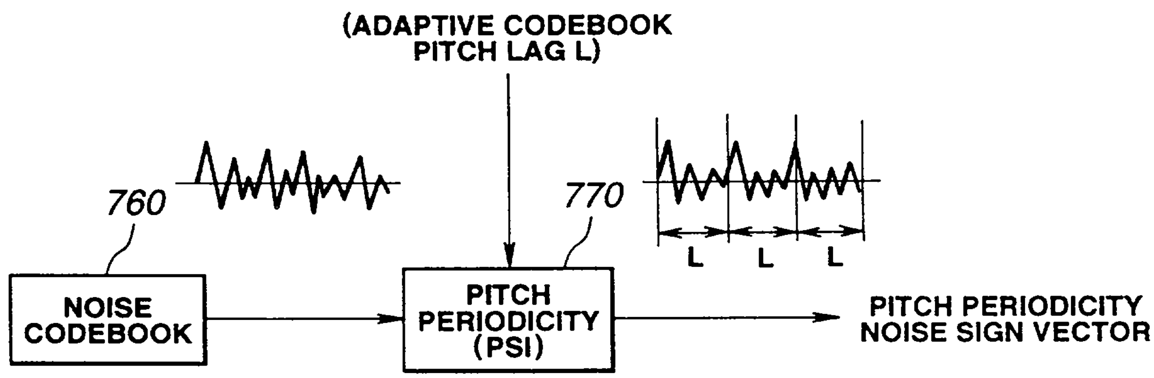
(ADAPTIVE CODEBOOK
PITCH LAG L)
760
770
L L L
NOISE CODEBOOK
PITCH PERIODICITY (PSI)
PITCH PERIODICITY NOISE SIGN VECTOR

FIG.7

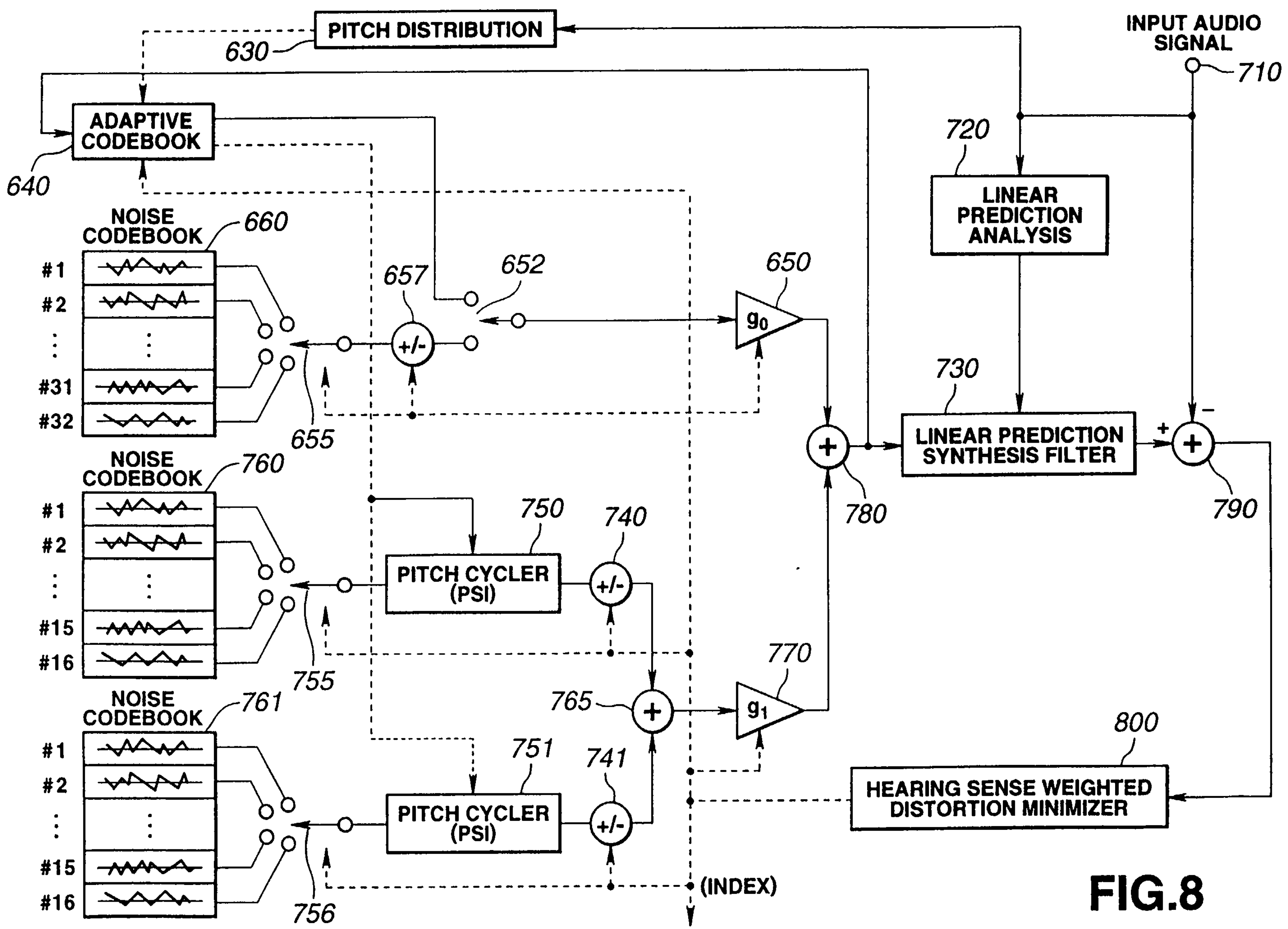
PITCH DISTRIBUTION
630
INPUT AUDIO SIGNAL
710
ADAPTIVE CODEBOOK
640
720
LINEAR PREDICTION ANALYSIS
NOISE CODEBOOK
660
#1
#2
#31
#32
657
652
650
g_0
655
730
LINEAR PREDICTION SYNTHESIS FILTER
780
790
NOISE CODEBOOK
760
#1
#2
#15
#16
750
PITCH CYCLER (PSI)
740
755
770
g_1
765
NOISE CODEBOOK
761
#1
#2
#15
#16
751
PITCH CYCLER (PSI)
741
756
800
HEARING SENSE WEIGHTED DISTORTION MINIMIZER
(INDEX)

FIG.8

S98P297US00
7217/55493

DECLARATION AND POWER OF ATTORNEY

As a below-named inventor, I hereby declare that:

My residence, post office address, and citizenship are as stated below next to my name.

I believe I am the original, first and sole inventor (if only one name is listed below) or an original, first and joint inventor (if plural names are listed below) of the subject matter which is claimed and for which a patent is sought on the invention entitled:

VECTOR SEARCH METHOD

the specification of which
(check one)

_X_ is attached hereto.

____ was filed on ____________________ as

Application Serial No. ____________________

and was amended on ____________________
(if applicable)

I hereby state that I have reviewed and understand the contents of the above-identified specification, including the claims, as amended by any amendment referred to above.

I acknowledge the duty to disclose information of which I am aware which is material to the examination of this application in accordance with Title 37, Code of Federal Regulations, Section 1.56(a).

I hereby claim foreign priority benefits under Title 35, United States Code, Section 119 of any foreign application(s) for patent or inventor's certificate listed below and have also identified below any foreign application for patent or inventor's certificate having a filing date before that of the application on which priority is claimed:

Prior Foreign Application(s)

Number	Country	Filing Date	Priority Claimed Yes	Priority Claimed No
P09-078615	Japan	March 28, 1997	X	

S98P297US00
7217/55493

I hereby claim the benefit under Title 35, United States Code, Section 120 of any United States Application(s) listed below and, insofar as the subject matter of each of the claims of this application is not disclosed in the prior United States application in the manner provided by the first paragraph of Title 35, United States Code, Section 112, I acknowledge the duty to disclose material information as defined in Title 37, Code of Federal Regulations, Section 1.56(a) which occurred between the filing date of the prior application and the national or PCT international filing date of this application:

Application Serial No.	Filing Date	Status

And I hereby appoint Jay H. Maioli, Reg. No. 27,213; Donald S. Dowden, Reg. No. 20,701; William E. Pelton, Reg. No. 25,702; Peter J. Phillips, Reg. No. 29,691; Gerald W. Griffin, Reg. No. 18,886; Ivan S. Kavrukov, Reg. No. 25,161; Christopher C. Dunham, Reg. No. 22,031; Norman H. Zivin, Reg. No. 25,385; John P. White, Reg. No. 28,678; and Robert D. Katz, Reg. No. 30,141; and each and all of them, all c/o Cooper & Dunham, 1185 Avenue of the Americas, New York, NY 10036 (Tel. (212) 278-0400), my attorneys, each with full power of substitution and revocation, to receive the patent, to transact all business in the Patent and Trademark Office connected therewith and to file any International Applications which are based thereon under the provisions of the Patent Cooperation Treaty.

Please address all communications, and direct all telephone calls, regarding this application to

JAY H. MAIOLI Reg. No. 27,213
Cooper & Dunham LLP
1185 Avenue of the Americas
New York, New York 10036
Tel. (212) 278-0400

I hereby declare that all statements made herein of my own knowledge are true and that all statements made on information and belief are believed to be true; and further that these statements were made with the knowledge that willful false statements and the like so made are punishable by fine or imprisonment, or both, under Section 1001 of Title 18 of the United States Code and that such willful false statements may jeopardize the validity of the application or any patent issued thereon.

Full name of sole or First joint inventor Yuji Maeda

Inventor's signature ____________________

Citizenship Japanese Date of Signature ____________________

Residence Tokyo, Japan

Post Office Address c/o Sony Corporation, 7-35 Kitashinagawa, 6-chome
Shinagawa-ku, Tokyo 141, JAPAN

S98P297US00
7217/55493

Full name of
Second joint inventor Shuichi Maeda

Inventor's signature ____________________

Citizenship Japanese **Date of Signature** ____________

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Shinagawa-ku, Tokyo 141, JAPAN